Amendments to the Claims

Please amend Claims 1, 29, 34, and 62. Please add new Claims 67 and 68. The Claim Listing below will replace all prior versions of the claims in the application:

Claim Listing

1. (Currently Amended) In a communication system for transmitting digital signals using a compression code comprising a predetermined plurality of parameters including a first parameter, said parameters representing an audio signal, said audio signal having a plurality of audio characteristics including a noise characteristic, said compression code being decodable by a plurality of decoding steps, apparatus for managing the noise characteristic comprising:

~~a receiver receiving digital signals, wherein said digital signals use a compression code comprising a predetermined plurality of parameters including a first parameter, wherein said parameters represent an audio signal,~~

~~wherein said audio signal has a plurality of audio characteristics including a noise characteristic,~~

~~wherein said compression code being decodable by a plurality of decoding steps; and~~

a processor responsive to said compression code of said digital signals to read at least said first parameter, and responsive to said compression code and said first parameter to generate an adjusted first parameter in a presence of speech, noise, and combination thereof and to replace said first parameter with said adjusted first parameter [[,]]

~~wherein said processor performs said plurality of decoding steps by performing first decoding steps to generate first decoder signals resulting in a noisy speech signal and second decoding steps to generate second decoder signals resulting in an estimated clean speech signal, and~~

~~wherein said processor responds at least to said first decoder signals and said second decoder signals and said first parameter to generate said adjusted first parameter.~~

2. (Canceled)

3. (Previously presented) Apparatus, as claimed in claim 1, wherein said first parameter comprises codebook gain, and wherein said processor modifies said codebook gain to modify a codebook vector contribution to said noise characteristic.

4. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises codebook gain, wherein said plurality of parameters further comprises pitch gain, wherein said plurality of characteristics further comprises signal to noise ratio and wherein said processor is responsive to said codebook gain, said pitch gain and said signal to noise ratio to generate said adjusted first parameter, and wherein said adjusted first parameter comprises an adjusted codebook gain.

5. (Original) Apparatus, as claimed in claim 4, wherein said signal to noise ratio comprises a ratio involving noisy signal power and noise power of said audio signal.

6. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises pitch gain, wherein said plurality of parameters further comprise codebook gain, wherein said processor performs said plurality of decoding steps by generating a codebook vector, wherein said processor scales said codebook vector by said codebook gain to generate a scaled codebook vector,

wherein said processor comprises at least a first buffer responsive to said scaled codebook vector to generate first samples based on pitch period, wherein said processor scales said first samples by said pitch gain to generate first scaled samples, and wherein said processor modifies said pitch gain to modify the contribution of said first scaled samples in order to manage said noise characteristic.

7. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises pitch gain, wherein said plurality of characteristics further comprises signal to noise ratio, wherein said processor is responsive to said pitch gain and said signal to noise ratio to generate said adjusted first parameter, and wherein said adjusted first parameter comprises an adjusted pitch gain.

8. (Original) Apparatus, as claimed in claim 7, wherein said signal to noise ratio comprises a ratio involving noisy signal power and noise power of said audio signal.

9. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises pitch gain, wherein said plurality of parameters further comprise codebook gain, wherein said processor performs said plurality of decoding steps to generate a codebook vector, wherein said processor scales said codebook vector by said codebook gain to generate a scaled codebook vector, wherein said processor generates a power signal representing the power of said scaled codebook vector, wherein said processor is responsive to said pitch gain and said power signal to generate said adjusted first parameter, and wherein said adjusted first parameter comprises an adjusted pitch gain.

10. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises pitch gain, wherein said processor comprises at least a first buffer generating at least first samples

based on pitch period, wherein said processor scales said first samples by said pitch gain to generate at least first scaled samples, wherein said processor generates at least a first power signal representing the power of said first scaled samples, and wherein said processor is responsive at least to said pitch gain and said first power signal to generate said adjusted first parameter, and wherein said adjusted first parameter comprises an adjusted pitch gain.

11. (Original) Apparatus, as claimed in claim 10, wherein said processor comprises a second buffer responsive in part to said first power signal to generate second samples based on pitch period, wherein said processor scales said second samples by said pitch gain to generate second scaled samples, wherein said processor generates a second power signal representing the power of said second scaled samples and wherein said processor is responsive to said pitch gain, said first power signal and said second power signal to generate said adjusted first parameter.

12. (Original) Apparatus, as claimed in claim 11, wherein said first buffer and said second buffer each comprises a long-term predictor buffer.

13. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises pitch gain, wherein said plurality of parameters further comprises a codebook gain, wherein said processor comprises a pitch synthesis filter, wherein said processor performs said plurality of decoding steps to generate a first vector, wherein said processor scales said first vector by said codebook gain to generate a scaled codebook vector, wherein said processor filters said scaled codebook vector through said pitch synthesis filter to generate a second vector, wherein said processor generates a power signal representing the power of said second vector, wherein said

processor is responsive to said pitch gain and said power signal to generate said adjusted first parameter, and wherein said adjusted first parameter comprises an adjusted pitch gain.

14. (Previously presented) Apparatus, as claimed in claim 13, wherein said first vector comprises a codebook excitation vector and wherein said second vector comprises a linear predictive coding excitation vector.

15. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises a codebook vector comprising pulses using variable sets of amplitudes, wherein said processor analyzes said sets to identify the powers of said noise characteristic represented by said sets, wherein said processor identifies a first set representing a power less than the power represented by said sets other than said first set, and wherein said processor adjusts said pulses according to said first set to generate said adjusted parameter.

16. (Original) Apparatus, as claimed in claim 1, wherein said plurality of decoding steps further comprises at least one decoding step that does not substantially affect the management of the noise characteristic and wherein said processor avoids performing said at least one decoding step.

17. (Original) Apparatus, as claimed in claim 16, wherein said at least one decoding step comprises post-filtering.

18. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises a linear predictive code.

19. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises regular pulse excitation – long term prediction code.

20. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises code-excited linear prediction code.

21. (Original) Apparatus, as claimed in claim 1, wherein said first parameter is a quantized first parameter and wherein said processor generates said adjusted first parameter in part by quantizing said adjusted first parameter before replacing said first parameter with said adjusted first parameter.

22. (Original) Apparatus, as claimed in claim 1, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said processor is responsive to said compression code to read at least said first parameter from each of said plurality of subframes, and wherein said processor replaces said first parameter with said adjusted first parameter in each of said plurality of subframes.

23. (Original) Apparatus, as claimed in claim 22, wherein said processor replaces said first parameter with said adjusted first parameter for a first subframe before processing a subframe following the first subframe to achieve lower delay.

24. (Original) Apparatus, as claimed in claim 1, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said processor begins to perform said decoding steps during a first of said subframes to generate a plurality of said decoded signals, reads said first parameter from a second of said subframes occurring subsequent to said first subframe, generates said adjusted first parameter in response to said decoded signals and said first parameter, and replaces said first parameter of said second subframe with said adjusted first parameter.

25. (Original) Apparatus, as claimed in claim 1, wherein said processor is responsive to said compression code to perform at least one of a plurality of said decoding steps to generate decoded signals and wherein said processor is responsive to said decoded signals and said first parameter to generate said adjusted first parameter.

26. (Previously presented) Apparatus, as claimed in claim 1, wherein said first parameter is selected from a group consisting of codebook vector, codebook gain, pitch gain and LPC coefficients representations, including line spectral frequencies and log area ratios.

27. (Previously presented) Apparatus, as claimed in claim 1, wherein said audio signals have spectral regions affected by said noise characteristic, wherein said first parameter comprises a representation of linear predictive coding coefficients, wherein said processor is responsive to said compression code and said representarion to determine said spectral regions affected by noise and to generate said adjusted first parameter to manage said noise characteristic in said regions, and

wherein said adjusted first parameter comprises an adjusted representation of linear predictive coding coefficients.

28. (Previously presented) Apparatus, as claimed in claim 27, wherein said representation of linear predictive coding coefficients is selected from a group consisting of line spectral frequencies and log area ratios.

29. (Currently Amended) In a communication system for transmitting digital signals comprising code samples, said code samples comprising first bits using a compression code and second bits using a linear code, said code samples representing an audio signal, said audio signal having a plurality of audio characteristics including a noise characteristic, apparatus for managing the noise characteristic comprising:

~~a receiver receiving digital signals comprising code samples, wherein said code samples comprise first bits using a compression code and second bits using a linear code,~~

~~wherein said code samples represents an audio signal,~~

~~wherein said audio signal has a plurality of audio characteristics including a noise characteristic; and~~

a processor responsive to said second bits to adjust said first bits and said second bits, without decoding said compression code, to manage ~~whereby~~ the noise characteristic in the digital signals ; and ~~is controlled,~~

a transmitter module to transmit adjusted first and second bits to a device to produce a corresponding audible signal with managed echo for an end user.

~~wherein said processor performs a plurality of decoding steps by performing first decoding steps to generate first decoder signals resulting in a noisy speech signal and second decoding steps to generate second decoder signals resulting in an estimated clean speech signal, and wherein said processor responds at least to said first decoder signals and said second decoder signals to generate said adjusted first bits.~~

30. (Original) Apparatus, as claimed in claim 29, wherein said linear code comprises pulse code modulation (PCM) code.

31. (Original) Apparatus, as claimed in claim 29, wherein said compression code samples conform to the tandem-free operation of the global system for mobile communications standard.

32. (Original) Apparatus, as claimed in claim 29, wherein said first bits comprise the two least significant bits of said samples and wherein said second bits comprise the 6 most significant bits of said samples.

33. (Original) Apparatus, as claimed in claim 32, wherein said 6 most significant bits comprise PCM code.

34. (Currently Amended) In a communication system for transmitting digital signals using a compression code comprising a predetermined plurality of parameters including a first parameter, said parameters representing an audio signal, said audio signal having a plurality of

audio characteristics including a noise characteristic, said compression code being decodable by a plurality of decoding steps, a method of managing the noise characteristic comprising:

reading at least said first parameter[[,]] ~~wherein said reading includes partially decoding said first parameter~~;

generating an adjusted first parameter in a presence of speech, noise, and combination thereof in response to said compression code and said first parameter; and

replacing said first parameter with said adjusted first parameter

~~performing said plurality of decoding steps by performing first decoding steps to generate first decoder signals resulting in a noisy speech signal and second decoding steps to generate second decoder signals resulting in an estimated clean speech signal; and~~

~~responding at least to said first decoder signals and said second decoder signals and said first parameter to generate said adjusted first parameter.~~

35. (Canceled)

36. (Previously presented) A method, as claimed in claim 34, wherein said first parameter comprises codebook gain, and wherein said method further comprises modifying said codebook gain to modify a codebook vector contribution to said noise characteristic.

37. (Original) A method, as claimed in claim 34, wherein said first parameter comprises codebook gain, wherein said plurality of parameters further comprises pitch gain, wherein said plurality of characteristics further comprises signal to noise ratio and wherein said generating comprises generating said adjusted first parameter in response to said codebook gain, said pitch gain

and said signal to noise ratio, and wherein said adjusted first parameter comprises an adjusted codebook gain.

38. (Original) A method, as claimed in claim 37, wherein said signal to noise ratio comprises a ratio involving noisy signal power and noise power of said audio signal.

39. (Original) A method, as claimed in claim 34, wherein said first parameter comprises pitch gain, wherein said plurality of parameters further comprise codebook gain, wherein said generating comprises performing said plurality of decoding steps by generating a codebook vector, scaling said codebook vector by said codebook gain to generate a scaled codebook vector, generating first samples based on pitch period in response to said scaled codebook vector, scaling said first samples by said pitch gain to generate first scaled samples, and modifying said pitch gain to modify the contribution of said first scaled samples in order to manage said noise characteristic.

40. (Original) A method, as claimed in claim 34, wherein said first parameter comprises pitch gain, wherein said plurality of characteristics further comprises signal to noise ratio, wherein said generating comprises generating said adjusted first parameter in response to said pitch gain and said signal to noise ratio, and wherein said adjusted first parameter comprises an adjusted pitch gain.

41. (Original) A method, as claimed in claim 40, wherein said signal to noise ratio comprises a ratio involving noisy signal power and noise power of said audio signal.

42. (Original) A method, as claimed in claim 34, wherein said first parameter comprises pitch gain, wherein said plurality of parameters further comprise codebook gain, wherein said generating comprises performing said plurality of decoding steps to generate a codebook vector, scaling said codebook vector by said codebook gain to generate a scaled codebook vector, generating a power signal representing the power of said scaled codebook vector, and generating said adjusted first parameter in response to said pitch gain and said power signal, and wherein said adjusted first parameter comprises an adjusted pitch gain.

43. (Original) A method, as claimed in claim 34, wherein said first parameter comprises pitch gain, wherein said generating comprises generating at least first samples based on pitch period, scaling said first samples by said pitch gain to generate at least first scaled samples, generating at least a first power signal representing the power of said first scaled samples, and generating said adjusted first parameter in response to at least said pitch gain and said first power signal, and wherein said adjusted first parameter comprises an adjusted pitch gain.

44. (Original) A method, as claimed in claim 43, wherein said generating further comprises generating second samples based on pitch period responsive in part to said first power signal, scaling said second samples by said pitch gain to generate second scaled samples, generating a second power signal representing the power of said second scaled samples and generating said adjusted first parameter in response to said pitch gain, said first power signal and said second power signal.

45. (Original) A method, as claimed in claim 44, wherein said system comprises one or more long-term predictor buffers and wherein said generating said first and second samples comprises using said one or more buffers.

46. (Original) A method, as claimed in claim 34, wherein said first parameter comprises pitch gain, wherein said plurality of parameters further comprises a codebook gain, and wherein said generating comprises performing said plurality of decoding steps to generate a first vector, scaling said first vector by said codebook gain to generate a scaled codebook vector, filtering said scaled codebook vector by pitch synthesis filtering to generate a second vector, generating a power signal representing the power of said second vector, and generating said adjusted first parameter in response to said pitch gain and said power signal, and wherein said adjusted first parameter comprises an adjusted pitch gain.

47. (Previously presented) A method, as claimed in claim 46, wherein said first vector comprises a codebook excitation vector and wherein said second vector comprises a linear predictive coding excitation vector.

48. (Original) A method, as claimed in claim 34, wherein said first parameter comprises a codebook vector comprising pulses using variable sets of amplitudes, wherein said generating comprises analyzing said sets to identify the powers of said noise characteristic represented by said sets, identifying a first set representing a power less than the power represented by said sets other than said first set, and adjusting said pulses according to said first set to generate said adjusted parameter.

49. (Original) A method, as claimed in claim 34, wherein said plurality of decoding steps further comprises at least one decoding step that does not substantially affect the management of the noise characteristic and wherein said generating avoids performing said at least one decoding step.

50. (Original) A method, as claimed in claim 49, wherein said at least one decoding step comprises post-filtering.

51. (Original) A method, as claimed in claim 34, wherein said compression code comprises a linear predictive code.

52. (Original) A method, as claimed in claim 34, wherein said compression code comprises regular pulse excitation – long term prediction code.

53. (Original) A method, as claimed in claim 34, wherein said compression code comprises code-excited linear prediction code.

54. (Original) A method, as claimed in claim 34, wherein said first parameter is a quantized first parameter and wherein said generating comprises generating said adjusted first parameter in part by quantizing said adjusted first parameter before replacing said first parameter with said adjusted first parameter.

55. (Original) A method, as claimed in claim 34, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said reading comprises reading at least said first parameter from each of said plurality of subframes in response to said compression code, and wherein said replacing comprises replacing said first parameter with said adjusted first parameter in each of said plurality of subframes.

56. (Original) A method, as claimed in claim 55, wherein said replacing comprises replacing said first parameter with said adjusted first parameter for a first subframe before processing a subframe following the first subframe to achieve lower delay.

57. (Original) A method, as claimed in claim 34, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said generating comprises beginning to perform said decoding steps during a first of said subframes to generate a plurality of said decoded signals, wherein said reading comprises reading said first parameter from a second of said subframes occurring subsequent to said first subframe, wherein said generating further comprises generating said adjusted first parameter in response to said decoded signals and said first parameter, and wherein said replacing comprises replacing said first parameter of said second subframe with said adjusted first parameter.

58. (Original) A method, as claimed in claim 34, wherein said generating comprises performing at least one of a plurality of said decoding steps to generate decoded signals in response

to said compression code and generating said adjusted first parameter in response to said decoded signals and said first parameter.

59. (Previously presented) A method, as claimed in claim 34, wherein said first parameter is selected from a group consisting of codebook vector, codebook gain, pitch gain and LPC coefficients representations, including line spectral pairs and line spectral frequencies.

60. (Previously presented) A method, as claimed in claim 34, wherein said audio signals have spectral regions affected by said noise characteristic, wherein said first parameter comprises a representation of linear predictive coding coefficients, and wherein said generating comprises determining said spectral regions affected by noise in response to said compression code and said representation and generating said adjusted first parameter to manage said noise characteristic in said regions, and wherein said adjusted first parameter comprises an adjusted representation of linear predictive coding coefficients.

61. (Previously presented A method, as claimed in claim 60, wherein said representation of linear predictive coding coefficients is selected from a group consisting of line spectral frequencies and log area ratios.

62. (Currently Amended) In a communication system for transmitting digital signals comprising code samples, said code samples comprising first bits using a compression code and second bits using a linear code, said code samples representing an audio signal, said audio signal

having a plurality of audio characteristics including a noise characteristic, a method of managing the noise characteristic comprising:

~~receiving digital signals comprising code samples, wherein said code samples comprise first bits using a compression code and second bits using a linear code,~~

~~wherein said code samples represent an audio signal,~~

~~wherein said audio signal has a plurality of audio characteristics including a noise characteristic;~~

adjusting said first bits and said second bits in response to said second bits, without decoding said compression code, to manage ~~whereby~~ the noise characteristic in the digital signals ~~is controlled,~~ ; and

transmitting adjusted first and second bits to a device to produce a corresponding audible signal with managed echo for an end user;

~~performing said plurality of decoding steps by performing first decoding steps to generate first decoder signals resulting in a noisy speech signal and second decoding steps to generate second decoder signals resulting in an estimated clean speech signal; and~~

~~responding at least to said first decoder signals and said second decoder signals and said first parameter to generate said adjusted first parameter.~~

63. (Original) A method, as claimed in claim 62, wherein said linear code comprises pulse code modulation (PCM) code.

64. (Original) A method, as claimed in claim 62, wherein said code samples conform to the tandem-free operation of the global system for mobile communications standard.

65. (Original) A method, as claimed in claim 62, wherein said first bits comprise the two least significant bits of said samples and wherein said second bits comprise the 6 most significant bits of said samples.

66. (Original) A method, as claimed in claim 65, wherein said 6 most significant bits comprise PCM code.

67. (New) Apparatus, as claimed in claim 1, wherein said processor performs said plurality of decoding steps by performing first decoding steps to generate first decoder signals resulting in a noisy speech signal and second decoding steps to generate second decoder signals resulting in an estimated clean speech signal, and wherein said processor responds at least to said first decoder signals and said second decoder signals and said first parameter to generate said adjusted first parameter.

68. (New) A method, as claimed in claim 34, and further comprising:

performing said plurality of decoding steps by performing first decoding steps to generate first decoder signals resulting in a noisy speech signal and second decoding steps to generate second decoder signals resulting in an estimated clean speech signal; and

responding at least to said first decoder signals and said second decoder signals and said first parameter to generate said adjusted first parameter.